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DIGITAL SELF-ADAPTING
GRAPHIC EQUALIZER AND METHOD

TECHNICAL FIELD OF THE INVENTION

This invention is related in general to the field of digital signal processing. More particularly, the invention is related to digital self-adapting graphic equalizer and method.

RELATED PATENT APPLICATION

This application is related to co-pending U.S. Serial No. 09/401,422, entitled *digital graphic equalizer control system and method* (Attorney Docket No. TI-28878), filed on September 22. 1999.

BACKGROUND OF THE INVENTION

The acoustic sound generated by conventional speakers contains a large degree of distortion due to the physical limitations of the mechanical structure of the speakers. For purposes of this document, "distortion" is used to mean sound pressure level variation as a function of frequency. Therefore, even with high quality audio processing and digital recording, the actual acoustic sound waves reproduced by conventional speakers are typically far from an accurate reproduction of the original sound. In low end to mid-range electronic consumer products, the distortion due to the speaker system is even more pronounced.

Self-adapting speaker equalization systems are presently costly to implement. These conventional self-adapting speaker equalization systems are therefore unsuitable to low end to mid-range electronic consumer products, such as lap-top computers and desktop computers. However, the demand for high quality sound in these systems is unprecedented due to the proliferation of multimedia applications, digital music download and playback, and world wide web multimedia websites.

SUMMARY OF THE INVENTION

Accordingly, there is a need for digital self-adaptive graphic equalizer system and method which produces good sound quality and are not costly to realize and implement as conventional systems.

In one aspect of the invention, a self-adaptive graphic equalizer is operable to equalize the affects of an audio system on an audio signal. The self-adaptive graphic equalizer includes an adaptive graphic equalizer having a plurality of equalizing filters, where the plurality of equalizing filters have different center frequencies equidistant from one another and spanning a predetermined audio bandwidth. Each equalizing filter is operable to filter an ith sub-band of the audio signal. A plurality of first filters are coupled to the audio system, each first filter is operable to filter an ith sub-band of an output signal of the audio system. A plurality of second filters are operable to filter an ith sub-band of the audio signal. A gain adjuster is operable to adjust the ith sub-band of the adaptive graphic equalizer in response to a difference in the ith sub-band of the filtered output signal from the plurality of first filters and the ith sub-band of the filtered audio signal from the plurality of second filters.

In another aspect of the invention, a digital self-adaptive graphic equalization method that equalizes the affects of a speaker-microphone system and the environment on an audio signal includes the steps of receiving an output signal from the audio system, the output signal being generated by the audio system in response to the audio signal. The output signal is divided into N sub-bands and an ith sub-band of the output signal is filtered, where $i = 1-N$. The audio signal is

also divided into the same N sub-bands and the ith sub-band of the audio signal is filtered, where i = 1-N. A difference between the ith filtered sub-band of the audio signal and the ith filtered sub-band of the output signal is determined, and the gain of an ith equalizing filter of an adaptive graphic equalizer is adjusted in response the difference between the ith filtered sub-band of the audio and output signals. The equalizing filters have different center frequencies equidistant from one another and spanning a predetermined audio bandwidth. An equalized audio signal is generated and provided to the audio system.

In yet another aspect of the invention, a digital self-adaptive graphic equalization method that equalizes the affects of a speaker-microphone system and the environment on an audio signal includes the steps of receiving an output signal from the audio system, the output signal being generated by the audio system in response to the audio signal. The output signal is divided into N sub-bands and an ith sub-band of the output signal is filtered, where i = 1-N. The audio signal is also divided into the same N sub-bands and the ith sub-band of the audio signal is filtered, where i = 1-N. The method time averages the N sub-bands of the filtered output signal, time averages the N sub-bands of the filtered audio signal, and normalizes the time averaged N sub-bands of the filtered output signal and the time averaged N sub-bands of the filtered audio signal. A difference between the ith filtered sub-band of the audio signal and the ith filtered sub-band of the output signal is determined, and the gain of an ith equalizing filter of an adaptive graphic equalizer is adjusted in response the difference between the ith filtered sub-band of the audio

and output signals. The equalizing filters have different center frequencies equidistant from one another and spanning a predetermined audio bandwidth. An equalized audio signal is generated and provided to the audio system.

BRIEF DESCRIPTION OF THE DRAWINGS

For a better understanding of the present invention, reference may be made to the accompanying drawings, in which:

FIGURE 1 is a simplified block diagram of a conventional adaptive equalizer;

FIGURE 2 is a simplified block diagram of an embodiment of a self-adaptive digital graphic equalizer with multiple filters each operating on a sub-band of the audio band constructed according to the teachings of the present invention;

FIGURES 3A and 3B are frequency plots of the filter sub-bands and the graphic equalizer filter family, respectively according to an embodiment of the present invention;

FIGURE 4 is a simplified block diagram of an embodiment of the ith band of self-adaptive digital graphic equalizer constructed according to an embodiment of the present invention; and

FIGURE 5 is a more detailed block diagram of an embodiment of an adaptive circuit and process of the self-adaptive digital graphic equalizer constructed according to an embodiment of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

FIGURE 1 is a simplified block diagram of a conventional adaptive equalizer 10. Digital self-adaptive graphic equalizer 10 receives an audio input signal 12, which is processed by an adaptive filter 14. The output from adaptive filter 14 is supplied to a speaker-microphone combination block 16. Speaker-microphone combination block 16 reproduces the audio signal processed by adaptive filter 14 and the acoustical sound output is measured by the microphone and provided as an acoustic input 18 to a summer 20. Summer 20 determines the difference between audio input 12 and acoustic input 18 and provides an adjustment input 22 to adaptive filter 14. Adjustment input 22 is used to adjust adaptive filter 14 to remove distortion introduced by the speaker, so that the difference between audio input signal 12 and the output from the speaker is minimized.

FIGURE 2 is a simplified block diagram of an embodiment of a self-adaptive digital graphic equalizer 30 with multiple filters each operating on a sub-band of the audio band constructed according to the teachings of the present invention. Self-adaptive digital graphic equalizer includes an adaptive graphic equalizer 34, which receives an audio input 32 and generates an output supplied to a speaker-microphone combination 36. The system can be pre-calibrated so that the unknown aspects of the performance are all due to the speaker and the play-back environment. For example, if the response of the microphone is not adequately flat, it can be pre-calibrated or equalized. If the microphone is relatively close to the speaker, the measured effects will be mostly due to the speaker itself, while the environment interaction becomes more apparent when the microphone is farther away from the speaker. Speaker-microphone

combination block 36 reproduces the audio signal processed by adaptive graphic equalizer 34 and the acoustical sound output is measured by the microphone and provided as an acoustic input to a filterbank 38. The

5   output from filterbank 38 is provided to a summer 42. A second filterbank 40 filters the audio input 32 and also provides output to summer 42. Accordingly, the audio band of interest is divided into a plurality of sub-bands and filtered by filterbanks 38 and 40. Summer 42

10   determines the difference between the filtered audio input and the filtered acoustic input measured by the microphone and provides an adjustment input 44 to adaptive graphic equalizer 34. Adjustment input 44 is used to adjust adaptive graphic equalizer 34 to remove

15   distortion introduced by the speaker, so that the difference between audio input signal 32 and the output from the speaker is minimized. FIGURE 3A contains frequency plots of a 10-band filterbank example, and FIGURE 3B is a frequency plot of the graphic equalizer

20   filter family example according to an embodiment of the present invention. For the sake of clarity, FIGURES 3A and 3B provides only the outlines of ±18 dB filters with fixed center frequency/bandwidth value, or Q, and fixed spacing in log-frequency space. It may be noted that the

25   range of the gain, the number of bands, the center frequencies, the spacing between filters, and the bandwidths, can be specified as desired for a particular application. Further, the spacing between the filters in each filter family and the center frequencies of the

30   filter families need not be equidistant. The filter gains can be adjusted to the desired level within the specified range. Self-adaptive graphic equalizer 30 can be used to automatically approximate an inverse function for the speaker in its environment by using the

35   difference between the outputs of the ith filters in

filterbanks 38 and 40 to adjust the gain of the ith filter
in the graphic equalizer family.

FIGURE 4 is a simplified block diagram of an
embodiment of the ith band of self-adaptive digital
graphic equalizer 30 constructed according to an
embodiment of the present invention. Audio input signal
32 is received by an ith band, G_i, 50 of graphic equalizer
34 (FIGURE 2), which produces an output that is then
supplied to the speaker of speaker-microphone combination
36. The microphone measures the acoustic output of the
speaker and provided as a measured acoustic input to an
i^{th} band, R_i, 52 of filterbank 38 applied to the measured
or received signal. Audio input 32 is supplied to an ith
band, O_i, 54 of filterbank 40 applied to the original
audio input signal. Typically, R_i and O_i are identical
filters. The ith band 50 of adaptive graphic equalizer 34
receives an ith adjustment signal from summer 56, which is
the difference between the filtered output of R_i and O_i
filters. Therefore, the ith band of graphic equalizer 34
is adjusted according to the difference from summer 56.

FIGURE 5 is a more detailed block diagram of an
embodiment of an adaptive circuit and process 60 of the
self-adaptive digital graphic equalizer constructed
according to an embodiment of the present invention. In
an embodiment of the present invention, filterbank 52
includes bandpass filters R_1 to R_N 61-63, which each
filter a sub-band of the measured acoustic output of the
speaker. The output from filters R_{1-N} 61-63 are provided
to low pass filters 65, LPF_{1-N} 66-68, which time average
the R_{1-N} filter outputs. Filterbank 54 includes bandpass
filters O_1 to O_N 90-92, which each filter a sub-band of
the original audio input signal 32. The output from
filters O_{1-N} 90-92 are provided to low pass filters 94,
LPF_{1-N} 95-97, which time average the O_{1-N} filter outputs.
Time averaging the filterbank outputs allows the

adaptation to be performed according to general trends rather than on instantaneous measurement values. Further, time averaging compensates for time delay differences between the two paths in the original audio signal and the measured acoustic output. The filtered original audio outputs, R_{Li}, are converted to dB space by 20 $\log_{10}$ circuits or algorithm 70-72. The filtered measured acoustic outputs, O_{Li}, are converted to dB space by 20 $\log_{10}$ circuits or algorithm 100-102. Accordingly,

$$r_{Li} = 20 \log_{10}(R_{Li}), and$$
$$o_{Li} = 20 \log_{10}(O_{Li})$$

The dB values, r_{Li} and o_{Li}, are provided to a mean normalization circuit or algorithm 76, which normalizes the values to compensate for differences in scaling the two paths. Either r_{Li} or o_{Li} can be manipulated such that

$$\sum_i r_{Li} = \sum_i o_{Li}$$

by using summers 80-82 and 104-106. The mean-adjusted r_{Li} and o_{Li} are represented as r'_{Li} or o'_{Li}, which are then provided to less than comparison circuit or algorithm 110-112. In one embodiment of the present invention, if $r'_{Li} < o'_{Li}$, then the gain in G_i is incremented by a predetermined amount; if $r'_{Li} < o'_{Li}$ is not true, then the gain in G_i is decremented by a predetermined amount. Alternatively, the adjustment in the gain of the appropriate sub-band of graphic equalizer 50 may be accomplished by using the difference between r'_{Li} and o'_{Li} as the gain in dB which the ith filter in the graphic equalizer is adjusted. As described in U.S. Application Serial No. 09/401422, entitled *digital graphic equalizer control system and method* (Attorney Docket TI-28878), if the gain adjustment is not performed gradually, undesirable audible artifacts results. This embodiment

allows for gradual adjustments in the gain to avoid audible artifacts.

Also discussed in *digital graphic equalizer control system and method*, a frequency of gain update more than every 64 samples of the audio input signal is likely to introduce audible artifacts. Moreover, a listener may be able to discern gain adjustment when it is done too frequently. Therefore, the frequency of gain adjustment should be no more than once per 64 samples, and possibly even slower.

In implementing the present invention, many parameters may be varied and determined based on the specific implementation, such as the number of sub-bands, the orders of the filters, the match between the shapes of the filterbanks and the graphic equalizer filters, center frequencies, maximum ranges, etc. For example, the Q values of the filters may be different and the center frequencies may have spacing other than octave spacing. Therefore, the system designer has the flexibility to tune and adjust the circuit or algorithm according to the application at hand.

Although performance is improved in general when a large number of filters are used, a simplification of using only one bandpass filter each cycle yields good results. In one embodiment, the audio information is passed through R_1 and O_1 for a cycle, the filtered audio information is then used to adjust the gain of G_1. R_2 and O_2 would then be applied for the next cycle and the filtered information is then used to update the gain of G_2.

Furthermore, overall convergence of the system is improved when each filter is adjusted individually than when all the filters are adjusted simultaneously. In addition, due in part to the overlapping of the sub-bands, more rapid convergence is achieved when non-

adjacent   sub-bands   are   adjusted   simultaneously.
Therefore in one embodiment, the full sets of bandpass
filters in the filterbanks are applied, but the gain
correction is only applied to one band each cycle.
Further, the adjustment may be done for sub-band i = 1
one cycle, i = 2 the next cycle, and so forth.
Alternatively, the sub-bands are adjusted by cycling
through the sub-bands every other or every third sub-
band, for example.   If fewer than the total number of
sub-bands will be adjusted in any one cycle, it is not
necessary to actually implement the bandpass filters
whose outputs are not needed.   In digital systems, it is
a simple matter to change the coefficients as needed,
allowing only the particular bandpass filter(s) needed to
be implemented.

In one embodiment, ten filter bands are used.   The
graphic equalizer filters are conventional second order
bell-shaped equalizer filters with Q = 2.   These filters
are centered approximately at the frequencies of C's on
the piano keyboard with two additional octaves added to
cover the entire audio band: 32.7, 65.4, 130.8, 261.6,
523.3, 1046.5, 2093.0, 4186.0, 8372.0, and 16744.0 Hertz.
The bandpass filters in both filterbanks may be second
order Butterworth filters designed to be centered at the
same frequencies as the corresponding graphic equalizer
filters.   As described in co-pending application *digital
graphic equalizer control system and method,* rather than
changing the equalizing filter coefficients such that the
filter gains change in uniform steps on a dB plot, the
coefficients may be changed such that the filter gains
change in uniform steps on a linear plot.   In this
manner, when the equalizing filter is at a high gain
setting, the gain is changed in very small and gradual
increments to avoid generating audible artifacts.   As the
gain decreases, it becomes possible to change the gain in

larger steps and yet still remain artifact free. In other words, by changing the coefficients such that the gain changes linearly, as opposed to logarithmically as in the traditional dB scale, artifacts would be eliminated. For example, 128 filters equally spaced in linear (gain) space between +/- 18 dB may be used. In this example, the linear spacing between the filters is approximately 0.0611.

The time-averaging lowpass filters may also be implemented in a variety of ways, including computing a simple running average. Alternatively, a simple alpha filter

$$y(n) = \alpha x(n) + (1 - \alpha) y(n-1)$$

with $\alpha = 2^{-11}$ may be used, for example.

As discussed in U.S. Patent Serial No. *09/407652*, .entitled *digital Audio Dynamic Range Compressor and Method*, (Attorney Docket TI-26912), the 20 $\log_{10}$ operation may be substituted by a base 2 logarithmic operation, which allows simple estimates. Therefore, the conversion to dB space may be performed by or replaced with base 2 logarithm estimates.

In operation, the graphic equalizer filter gains adjust over time and improves the performance of the system. Alternatively, the system may be pre-calibrated by first processing a white noise signal as input, to allow the filters to converge prior to operating on the actual audio signals. Thereafter, the graphic equalizer filter gains can be fixed or allowed to continue to adapt.

According to teachings of the present invention, the division of the audio band into various sub-bands may be extended beyond speaker equalization and be applied to a general inverse filtering problem, where an unknown system alters the audio signal. The present invention

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may be implemented to pass the altered signal through a filterbank of N filters and adjust the gain of the corresponding sub-band of the graphic equalizer according to a difference between the same sub-band of the filtered input signal. Furthermore, although the present invention is shown and described as being applied to a single channel, a multi-channel system would similarly benefit from the application of the present invention.

Although several embodiments of the present invention and its advantages have been described in detail, it should be understood that mutations, changes, substitutions, transformations, modifications, variations, and alterations can be made therein without departing from the teachings of the present invention; the spirit and scope of the invention being set forth by the appended claims.